REMARKS

Applicants express appreciation to the Examiner for consideration of the subject patent application. This amendment is in response to the Office Action mailed January 11, 2007. Claims 1-11 and 14-59 were rejected. The claims have been amended to address the concerns raised by the Examiner.

Claims 1-11 and 14-59 were originally presented. Claims 1-11 and 14-59 remain in the application. Claims 26-28 have been canceled without prejudice. Claims 1, 4-11, 14-25, 30-38, 41, 44, 46-51, 53, and 55-58 have been amended. No claims have been added.

Claim Rejections - 35 U.S.C. § 112

Claims 30-45 stand rejected under § 112, 2nd paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention. Independent claim 30 has been amended to remove the phrase "can give rise." Independent claim 38 has been amended to address an issue relating to antecedent basis .

Claim Rejections - 35 U.S.C. § 103

Claims 1-11, 14-16, and 18-59 (including independent claim 1, 20, 30, 38, 46, and 53) were rejected under 35 U.S.C. § 103(a) as being unpatentable over Tibbetts (4,056,742) in view of Tanaka et al. (4,823,908). The claims have been amended to clarify that the film diaphragm recited in the claims is a polymer film diaphragm. Thus, the claimed invention includes the limitation of a <u>polymer film</u> diaphragm for use in producing parametric sound. Applicants respectfully submit that there is no motivation to combine the film transducer of Tibbetts and the parametric loudspeaker of Tanaka to arrive at the presently claimed invention. To that end, a brief history of industry progress in parametric sound production follows.

Prior to the present application, parametric speaker systems were constructed using a large array of small, piezoelectric bimorph ceramic transducers. There are a number of positive attributes that have led those skilled in the art to use the bimorph transducers. The transducers are small and can be relatively energy efficient if they are used at their resonant frequency. The bimorph transducers are off-the-shelf components that typically exhibit a high sound pressure level (SPL) output. The high SPL of the bimorph transducers can help to overcome the relatively

large inefficiencies in the coupling of ultrasonic parametric waves to air and demodulation of the parametric waves to produce audible sound. Therefore, the understanding of those skilled in the relevant art prior to the present invention was that more power injected into the air in the output of the parametric speaker would result in better quality sonic sound.

Prior to the present invention, those skilled in the relevant art sought to achieve better quality sonic output from a parametric speaker by using larger and larger arrays of high power bimorph ceramic transducers to increase the audible sound that decoupled from the parametric waves. For example, the parametric loudspeaker of Tanaka uses an array of 120 separate piezoelectric transducers arranged in a honeycomb pattern on a substrate. (Col. 7, lines 3-8). Tanaka does not disclose the physical properties of the piezoelectric transducers disclosed in the patent. However, the limitations described in the specification, specifically that each transducer has a 9.7 mm diameter, a center frequency (aka resonant frequency) of 40 kHz, and a sound pressure level of 123 dB, correspond to the specifications of a ceramic bimorph transducer. Therefore, one skilled in the relevant art would conclude that Tanaka uses an array of bimorph transducers. Later attempts to produce a commercially successful parametric loudspeaker included increasingly larger numbers of ceramic transducers (up to 2000) to achieve a desired volume after the sonic signal was decoupled from the ultrasonic parametric waves output from the array.

Articles supporting this assertion are included in an appendix to this Office Action response. Exhibit A of the appendix discloses an "audio spotlight" that is a parametric speaker formed using 547 PZT bimorph transducers. (See Exhibit A, Masahide Yoneyama, Jun-ichiroh Fujimoto, Yu Kawamo, Shoichi Sasabe "*Audio Spotlight: An Application of Nonlinear Interaction of Sound Waves to a New Type of Loudspeaker Design*" J. Acoustical Society of America 73(5), May 1983, Pages 1532-1536, p. 1534, Section II). Moreover, Exhibit B of the appendix discloses a parametric loudspeaker that includes 1410 piezoelectric transducers with a diameter of 1 cm and a resonant frequency of 28 kHz. (See Exhibit B, Kenichi Aoki, Tomoo Kamakura, Yoshiro Kumamoto "*Parametric Loudspeaker – Characteristics of Acoustic Field and Suitable Modulation of Carrier Ultrasound*" Electronics and Communications in Japan, Part 3, Vol. 74, No. 9, 1991, pages 76-80, p. 78, 2nd paragraph). Additionally, Exhibit C of the appendix discloses a parametric loudspeaker that includes "about 2000 small PZT bimorph

transducers of resonant frequency 28 kHz." (See Exhibit C, Tomoo Kamakura, Kenichi Aoki, Yoshiro Kumamoto "*Suitable Modulation of the Carrier Ultrasound for a Parametric Loudspeaker*" Acustica, Vol. 73, 1991, pages 215-217, p. 217, 1st paragraph).

Thus, the articles disclosing parametric speakers that existed prior to the present invention illustrate that, prior to the present invention those skilled in the relevant art sought to increase parametric speaker sound pressure level (SPL) output by using increasingly larger arrays of bimorph transducers in an attempt to leverage the properties of the bimorphs, as previously discussed, to increase the SPL to a level where audible sound having desired volume could be decoupled from the parametric output.

The properties of polymer film transducers, such as the polymer film diaphragm in the present claims, are substantially different from typical cone-type speakers and ceramic bimorph transducers that had been used in the prior art. For example, one important aspect of polymer film transducers in general is that they typically are not capable of producing sonic or ultrasonic waves with a sound pressure level that is near the amplitude achieved with more traditional transducers, such as the ceramic bimorph transducers used in parametric sound systems at the time.

Tibbetts does not disclose the sound pressure level at which the piezoelectric transducer can produce ultrasonic waves. However, US 4,246,448 to Tam et al. discloses an electrostatic transducer designed for use as a loudspeaker that is capable of producing a maximum sound pressure level of 105 dB. (See Tam, Column 8, lines 60-62). **The output of the electrostatic transducer loudspeaker is 18 dB (63 times) lower than *each* of the 120 transducers disclosed in Tanaka**. Similar or lower outputs were typical at the time of the invention for other types of film transducers, such as the piezoelectric film transducer of Tibbetts.

Therefore, **one skilled in the art would be significantly deterred from using the relatively low power piezoelectric film transducer of Tibbetts in place of the array of higher power bimorph transducers disclosed in Tanaka**. One skilled in the art would not have turned to the use of thin film diaphragms when it was understood that higher power (greater SPL) was needed to overcome the deficiencies of parametric loudspeakers. This is supported by the cited articles included in the appendix that show arrays of up to 2000 bimorph transducers used to increase the sound pressure level of parametric loudspeakers.

However, the present inventors discovered significant, unexpected results when using polymer film transducers instead of an array of bimorph ceramic transducers. Despite the fact that polymer film transducers output dozens of times less power per unit area than an array of bimorph transducers, it was found that the polymer film transducer was capable of overcoming many of the obstacles that had previously limited the commercial potential of parametric speaker systems.

Notwithstanding the many positive attributes of bimorph transducers, as previously discussed, the inventors found that bimorph transducers also had several drawbacks that contributed to the lack of marketability of parametric sound systems at the time. For example, the ability of bimorph transducers to output very high power acoustic waves was considered favorably by those skilled in the art due to the need to output parametric waves having an extremely high SPL to produce a decoupled sonic wave having sufficient power, as previously discussed. However, the present inventors discovered that high drive intensity immediately in front of each bimorph device can readily drive the air into shock or saturation. This phenomenon can break down the effective demodulation of the audio signal, causing loss of power output and severe distortion of the audio sound component, as well as other serious adverse effects upon the general process of parametric loudspeaker operation. In addition, bimorph transducers have poor frequency response and unwanted sub-harmonics.

While it was perceived at the time of the invention that increasing the number of bimorph emitters would provide increased SPL output in the ultrasonic range, it was also discovered that the use of a large number of bimorph emitters can merely exaggerate the problem of air saturation and serious power loss. Furthermore, the inventors have discovered a number of accompanying limitations with phase matching errors due to variations from device to device in the bimorph emitters, along with distortion and bandwidth problems and the associated cost and complexity of using so many separate devices. Indeed, it was discovered that the phase relationships of these separate devices are such that the total output of many devices used as a cluster does not equal the amount that would be predicted by summing the output of all of the devices.

The use of polymer film diaphragms, as recited in the claims of the present application, was found to unexpectedly overcome many of the limitations of an array of bimorph transducers.

Specifically, the lower power per area of a polymer film diaphragm does not require driving the air into shock or saturation to produce audible sound from a parametric wave, as is typically done with bimorph transducers. This allows the sonic signal to demodulate from the ultrasonic carrier signal more effectively, with less loss. Additionally, even when a film diaphragm is divided into a plurality of sections, the sections can be more homogenous and more accurately controlled with respect to phase to allow the output of the transducer to have better phase characteristics, creating an acoustic signal with greater output and better sound quality. Moreover, although the film diaphragm has lower SPL output per unit area than the bimorph transducers, the ability to drive the monolithic film to produce an output that is substantially in phase can provide greater overall power output than is possible with the hundreds or thousands of bimorph transducers that each have disparate phase.

Furthermore, the ability to better control the phase characteristics of the parametric waves produced using polymer film diaphragms has enabled additional advances in the field of parametric sound production. For example, signal processing algorithms have been developed for parametric sound systems to pre-compensate for distortion that occurs in the demodulation of the parametric pressure waves into audible sound. The function of the signal processing algorithms is inherently superior when the phase of the parametric pressure waves can be more closely controlled, as occurs when using large area polymer film diaphragm(s). Even if the polymer film diaphragms are driven at a power sufficient to drive the air into saturation, the ability to more closely control the phase output of the waves emitted from the polymer film diaphragms enables the use of signal processing algorithms that allow for pre-compensation to substantially reduce distortion in the audible sound.

In contrast, the use of hundreds or thousands of separate bimorph transducers creates a parametric pressure wave with hundreds and thousands of different phases due to the physical differences in each of the bimorph transducers. The different phases in the parametric pressure waves can significantly impede any reduction in distortion using the signal processing algorithms.

Thus, the use of a polymer film diaphragm rather than an array of bimorph transducers stands in contrast to the teachings of Tanaka. The Tanaka and Tibbetts references teach away from each other and the claimed invention. Tanaka discloses the use of a plurality of high power

piezoelectric bimorph transducers. Tibbetts discloses a single emitter having substantially less power than the emitters in Tanaka. Those skilled in the relevant art at the time the present invention was made would not have resorted to the use of lower power emitters when all of the art taught that greater power was needed.

The inventors' use of film diaphragms went against the teachings of the art at the time of the invention. Rather than using devices that produce greater sound pressure levels, as disclosed in Tanaka, the inventors discovered that they can achieve better audio quality in parametric sound systems by using lower power polymer film diaphragms that provide unexpected advantages over the higher power bimorphs, such as a greater ability to control a phase of the parametric waves emitted from the film diaphragms. The use of the polymer diaphragms has led to additional discoveries, such as the pre-compensation algorithms, that may not have been possible if the inventors had followed conventional wisdom taught in the prior art and used a large array of separate bimorph transducers, each with varying phases and power outputs due to differences in construction.

The Office Action states that one skilled in the art would review the piezoelectric film transducer structure disclosed in Tibbetts and search for related art on how to drive the piezo transducer. However, the Tibbetts reference does not disclose an ultrasonic transducer, nor does it disclose a large area transducer, as recited in the claims of the present invention. There would be no motivation or reason for one skilled in the art to refer to the Tibbetts reference at all since the prior art in parametric sound production relied on large arrays of small, bimorph transducers as referenced in the appendix.

Therefore, Applicants respectfully submit that there is no motivation to combine the Tibbetts and Tanaka references and claims 1-11, 14-16, and 18-59 are allowable. In light of the above discussion and the support in the appendix, Applicants urge the Examiner to withdraw the rejection.

Claim 17 was rejected under 35 U.S.C. § 103(a) as being unpatentable over Tibbetts in view of Tanaka as applied to claim 1 above, and further in view of Sakagami et al. (4,784,915).

Rejection of dependent claim 17 should be reconsidered and withdrawn for at least the reasons given above with respect to the independent claims. The dependent claim, being

narrower in scope, is allowable for at least the reasons for which the independent claims are allowable.

Double Patenting

Claims 1, 20, 30, 38, 46, and 53 stand provisionally rejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over claim 28 of copending Application No. 10/923,295 in view of Tanaka et al. (4,823,908). Applicants acknowledge the provisional rejection and will consider the filing of a terminal disclaimer to address this issue should the need arise in the future.

Claims 1, 20, 30, 38, 46, and 53 stand provisionally rejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over claim 32 of copending Application No. 10/923,295 in view of Tanaka et al. (4,823,908). Applicants acknowledge the provisional rejection and will consider the filing of a terminal disclaimer to address this issue should the need arise in the future.

CONCLUSION

In light of the above, Applicants respectfully submit that pending claims 1-11 and 14-59 are in condition for allowance. Therefore, Applicants request that the rejections and objections be withdrawn, and that the claims be allowed and passed to issue. If any impediment to the allowance of these claims remains after entry of this Amendment, the Examiner is strongly encouraged to call Alex W. Haymond at (801) 566-6633 so that such matters may be resolved as expeditiously as possible.

Fees in the amount of $510.00 will be submitted electronically pursuant to 37 C.F.R. § 1.17(a)(3), for a 3 month extension of time pursuant to 37 C.F.R. § 1.136. No claims were added; therefore, it is believed that no additional fee is due.

The Commissioner is hereby authorized to charge any additional fee or to credit any overpayment in connection with this Amendment to Deposit Account No. 20-0100.

DATED this 11th day of July, 2007.

Respectfully submitted,

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Appendix

Exhibit A

The audio spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design

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This work was done to devise a new type of loudspeaker. The theory for sound reproduction of this loudspeaker is based on nonlinear acoustics of sound wave interaction in air. A finite amplitude ultrasound wave that can be amplitude modulated by any audio signal is radiated from a transducer array into air as the primary wave. As a result, an audio signal is produced in the air because of the self-demodulation effect of the AM sound wave due to the nonlinearity of the air. It is possible to get a flat characteristic of reproduced sound pressure by using an equalizer. In some fundamental experiments the characteristic of the reproduced sound pressure is not quite flat due to an imperfect transducer array. Improvement of the transducer makes it possible to get a flat characteristic. A special feature of this loudspeaker is its very sharp directivity pattern, which makes it possible to realize a sound spotlight.

PACS numbers: 43.25.Lj, 43.25.Vt, 43.88.Ja

INTRODUCTION

The nonlinear interaction of finite amplitude ultrasonic waves in the air can be applied to a loudspeaker.[1] This paper describes the fundamental concept of a loudspeaker based on the nonlinear interaction of sound waves in air. Also, some of the experimental results from the operation of a prototype loudspeaker are presented.

When two finite amplitude sound waves (primary waves), having different frequencies, interact with one another in a fluid, new sound waves (secondary waves) whose frequencies correspond to the sum and the difference of the primary waves may be produced as the result.

This phenomenon was first analyzed by Westervelt[2] and is well known as "nonlinear interaction of sound waves," or the "scattering of sound by sound."[3] Based on Lighthill's arbitrary fluid motion equation[4] as shown in Eq. (1), Westervelt derived an inhomogeneous wave equation which is satisfied by the sound pressure of secondary waves produced by the nonlinear interaction [Eq. (2)].

$$\frac{\partial^2 \rho}{\partial t^2} - c_0^2 \nabla^2 \rho = \frac{\partial^2 T_{ij}}{\partial x_i \partial x_j}, \tag{1}$$

ρ: density of fluid, T_{ij}: stress tensor,

$$\nabla^2 p_s - \frac{1}{c_0^2}\frac{\partial^2 p_s}{\partial t^2} = -\rho_0 \frac{\partial q}{\partial t},$$

$$q = \frac{\beta}{\rho_0^2 c_0^4}\frac{\partial}{\partial t} p_1^2. \tag{2}$$

In Eq. (2), p_s is the secondary wave sound pressure, p_1 is the primary wave sound pressure, β is the nonlinear fluid parameter, and c_0 is the small signal sound velocity.

The solution for Eq. (2) may be expressed by the superposition integral of the Green's function and the virtual second source [right side of Eq. (2)] as shown in Eq. (3).

$$p_s = \frac{\rho}{4\pi}\iiint_v \frac{1}{|\mathbf{r}-\mathbf{r}'|}\frac{\partial}{\partial t} q\left(\mathbf{r}, t - \frac{|\mathbf{r}-\mathbf{r}'|}{c_0}\right) d\mathbf{r}', \tag{3}$$

where $\mathbf{r}$ is the observation point position vector, $\mathbf{r}'$ is the source position vector and v is the nonlinear interaction space.

When the primary wave consists of two continuous sinusoidal waves and both are planar and well collimated, the integral of Eq. (3) is calculated in the same manner as in previous papers.[2,5] When the directivity of a circular piston is taken into consideration, however, Eq. (3) must be used with the expression of Muir *et al.*[6]

A new type of loudspeaker has been developed on the basis of the nonlinear interaction of sound waves mentioned above. In this type of loudspeaker, ultrasound is amplitude modulated by an audio signal and radiated from a transducer array as finite amplitude waves. When the amplitude-modulated ultrasound wave interacts is a nonlinear fashion in air, the modulated signal (the audio signal) can be demodulated in the air.[1]

In the following section, the principle underlying this type of loudspeaker is described.

I. THEORY

A. Acoustic reproduction by nonlinear interaction of AM ultrasound in air

When two sinusoidal sound waves are radiated in the air, two new waves with angular frequencies of $\omega_1 \pm \omega_2$ arise by nonlinear interaction of the two original sinusoidal waves, whose angular frequencies are ω_1 and ω_2.

Therefore one might expect the secondary wave which corresponds to the modulation signal, to appear in the air as a result of the nonlinear interaction between the carrier ultrasound and the lower and upper sideband waves, provided that a finite amplitude AM ultrasound wave is radiated into

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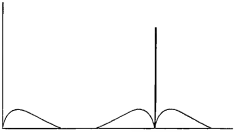

FIG. 1. Frequency spectra of an AM wave and demodulated wave.

the air. That is, the AM ultrasound is self-demodulated by the nonlinear interaction.

Figure 1 shows the spectra for both an AM wave and a demodulated wave. In this case, since the modulation wave is reproduced in the air, a new type of loudspeaker can be devised if the modulation signal is selected as the program audio signal.

If a finite amplitude ultrasound beam, modulated by an audio signal $g(t)$, is radiated into the air from a transducer array, the sound pressure p_1 of the primary wave (AM wave) at a distance x from the array on axis may be represented by Eq. (4)

$$p_1 = p_0[1 + mg(t - x/c_0)]e^{-\alpha x} \sin \omega_0(t - x/c_0), \quad (4)$$

where p_0 is the initial sound pressure of the ultrasound, m is the parameter indicating modulation index, and α is the absorption coefficient of carrier sound.

A virtual audio signal source occurs in the primary sound beam because of the nonlinearity of the acoustic interaction in air. This sound source may be represented by Eq. (5) using Eq. (2) and Eq. (4)

$$q = \frac{\beta p_0^2}{\rho_0^2 c_0^4} e^{-2\alpha x} \frac{\partial}{\partial t}\left[mg\left(t - \frac{x}{c_0}\right) + \frac{1}{2} m^2 g^2\left(t - \frac{x}{c_0}\right)\right]. \quad (5)$$

In the above equation, the second term on the right side implies a harmonic distortion component arising from the interaction between the lower and upper sideband waves. If the primary sound beam cross section is assumed to be circular with radius a, then the demodulated audio sound pressure p_s at the point r from the array, on axis, can be calculated analytically using Eqs. (3) and (5) in the form

$$p_s = \frac{\beta p_0^2 a^2 m}{8\rho_0 c_0^4 \alpha r} \frac{\partial^2}{\partial t^2} g\left(t - \frac{r}{c_0}\right). \quad (6)$$

On the other hand, the sound pressure of a harmonic distortion component may be expressed as

$$p_d = \frac{\beta p_0^2 a^2 m^2}{16\rho_0 c_0^4 \alpha r} \frac{\partial^2}{\partial t^2} g^2\left(t - \frac{r}{c_0}\right). \quad (7)$$

The Fourier transform of Eq. (6) can be expressed as

$$P_s(\omega) = -(\beta p_0^2 a^2 m / 8\rho_0 c_0^4 \alpha r)\omega^2 \exp[-j(r/c_0)\omega]\, G_s(\omega), \quad (8)$$

where $P_s(\omega)$ is the Fourier transform of $p_s(t)$, and $G_s(\omega)$ is the Fourier transform of $g(t)$. As evident from Eq. (8), $P_s(\omega)$ is proportional to ω^2 and thus the frequency characteristics of the reproduced sound show a 12 dB/oct dependence. Consequently, the audio signal (modulation signal) must be processed by an equalizer having -12 dB/oct frequency characteristics before the audio signal is introduced into the AM modulator.

B. Harmonic distortion

In the case of pure-tone modulation, $g(t) = \sin \omega t$, the sound pressures arising from both the signal secondary wave and the second harmonic distortion signal are calculated from Eqs. (6) and (7), respectively,

$$p_s(t) = -(\beta p_0^2 a^2 m \omega^2 / 8\rho_0 c_0^4 \alpha r) \sin \omega(t - r/c_0), \quad (9)$$

$$p_d(t) = (\beta p_0^2 a^2 m^2 \omega^2 / 8\rho_0 c_0^4 \alpha r) \cos 2\omega(t - r/c_0). \quad (10)$$

From these equations, it is possible to define the second harmonic distortion ratio as follows

$$\epsilon = [|p_d(t)|/|p_s(t)|] \times 100 = m \times 100\%. \quad (11)$$

Because the second harmonic distortion ratio is proportional to m, a good distortion ratio requires a very small modulation depth to prevent cross interaction between the lower and upper sideband waves. The signal and distortion sound waves are represented by the first and the second term on the right side of Eq. (5), respectively. The sound pressure of the signal is proportional to m, while the distortion is proportional to m^2. In accordance with this relation, if m is selected less than 1, the distortion sound pressure will be much less than the signal sound pressure.

If the equalizer of -12 dB/oct is used, the modulation depth m varies with the frequency of the modulation signal, as expressed in Eq. (12)

$$m = m_0/\omega^2, \quad m_0 \text{ is constant.}$$

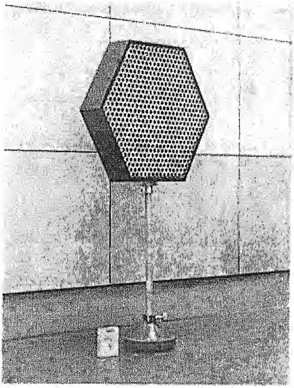

FIG. 2. Front view of the loudspeaker.

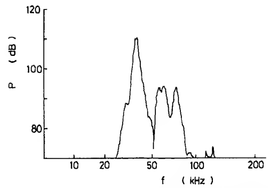

FIG. 3. Sound pressure-frequency response characteristics of the transducer array, for a point 4 m from the transducer. The input voltage is 0.5 V.

In this case, since the second harmonic distortion ratio ϵ is proportional to $1/\omega^2$, distortion in low-frequency regions increases markedly.

That ϵ is proportional to $1/\omega^2$, in spite of the flatness of the signal frequency characteristics, is due to the fact that $p_d(t)$ is proportional to m^2 even through $p_s(t)$ is proportional to m. If m is kept small to make distortion low, the sound pressure $p_s(t)$ also decreases.

Therefore, either the initial sound pressure p_0 of the carrier wave or the radius of the primary beam cross section should increase to maintain the expected $p_s(t)$.

II. EXPERIMENT

A loudspeaker using a finite amplitude AM ultrasound radiated from a transducer array was developed and put to practical use. This array consisted of 547 PZT bimorph transducers. The fundamental resonant frequency of each transducer was about 40 kHz. A front view of the array appears in Fig. 2.

The sound pressure frequency response characteristics

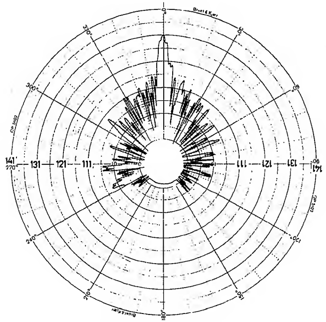
FIG. 4. Directivity at 40 kHz of the transducer array, for a point 4 m from the transducer. The input voltage is 10 V.

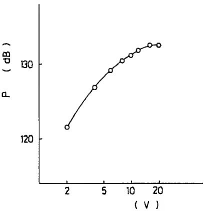

FIG. 5. Sound pressure versus input voltage at 40 kHz, for a point 4 m from the transducer.

and the directivity at 40 kHz (the primary wave) of the array are shown in Figs. 3 and 4, respectively. As can be seen from Fig. 3, the frequency response characteristics of the array are not symmetrical for 40 kHz. Moreover, there are many harmonic resonances and antiresonances. The frequency response characteristics of the secondary sound wave are distributed by the resonances and antiresonances.

Figure 5 shows the sound pressure at 40 kHz, at a point 4 m from the array, plotted against input voltage.

The sound pressure frequency response characteristics of the secondary wave produced by the nonlinear self-interaction of the finite amplitude AM ultrasound radiated from the array, are shown in Fig. 6. The characteristics were measured with modulation depth $m = 0.5$ at a point 4 m from the array in an anechoic chamber. The 12 dB/oct equalizer was not used. In the frequency region below 1.5 kHz, the characteristics almost follow the 12 dB/oct curve. The sound pressure characteristics of the primary wave have a flat region within the frequencies of 40 ± 1.5 kHz as shown in Fig. 3. When the sideband spectra of the modulated ultrasound deviates from the flat range, the sound pressure of the secondary wave decreases. The peak of the primary sound pressure curve at 60 kHz produces the peak of the secondary wave at 20 kHz. All of these phenomena can be predicted

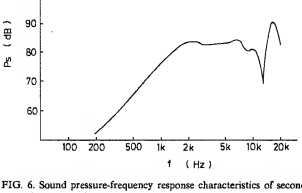

FIG. 6. Sound pressure-frequency response characteristics of secondary wave, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

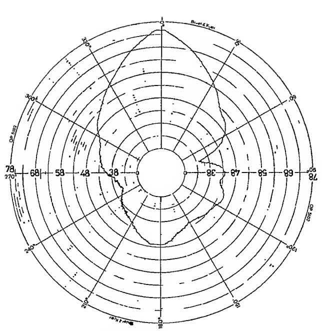

FIG. 7. Directivity of secondary wave at 1.0 kHz, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

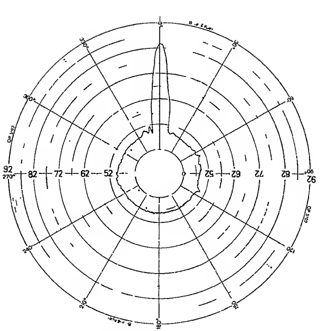

FIG. 9. Directivity of secondary wave at 10.0 kHz, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

from Eq. (9) and the characteristics of the primary wave.

The measured directivities of the secondary signal waves at 1.0, 5.0, and 10.0 kHz are shown in Figs. 7, 8, and 9, respectively.

To check the relation between the secondary signal sound pressure $p_s(t)$ and second harmonic component sound pressure $p_d(t)$ of the secondary wave, the secondary wave picked up by audio microphone was analyzed by a spectrum analyzer for various values of m. Figure 10 shows the measurement results at $f_s = 5.0$ kHz. These results show that the relation of the sound pressure level between signal and distortion are predicted by Eqs. (9) and (10). For example, if the results of $m = 1.0$ and $m = 0.5$ are compared, it is clear that the signal level (i.e., 5 kHz) decreases 6 dB and the second

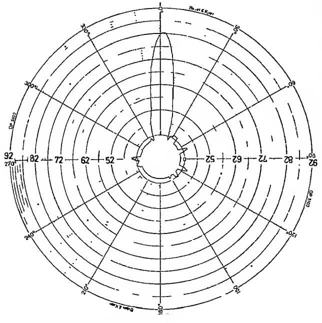

FIG. 8. Directivity of secondary wave at 5.0 kHz, for a point of 4 m, $m = 0.5$, and input voltage of 10 V.

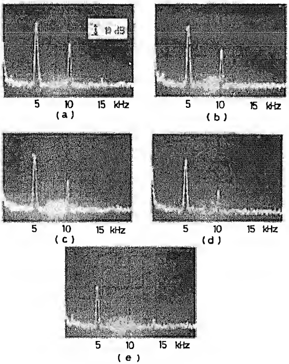

FIG. 10. Relations of secondary signal sound pressure p_s and second harmonic sound pressure p_d, (a) $m = 1.0$, (b) $m = 0.7$, (c) $m = 0.5$, (d) $m = 0.3$, and (e) $m = 0.1$.

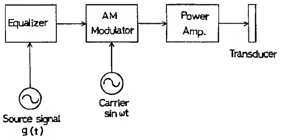

FIG. 11. Construction of the loudspeaker.

harmonic distortion level (10 kHz) decreases 12 dB. Accordingly, the signal sound pressure is proportional to m and that of the distortion is proportional to m^2.

III. DISCUSSION

An entirely new type of loudspeaker has been developed. This research is based on the phenomenon of the nonlinear interaction of sound waves. That is, the self-modulation effect of finite amplitude AM ultrasound by the nonlinearity of the air has been applied in the construction of the loudspeaker. This loudspeaker consists of an ultrasound transducer array, a driving amplifier for the array, an AM modulator, a pure-tone oscillator for the carrier frequency and equalizer as shown in Fig. 11.

The sound pressure obtained from the loudspeaker is proportional to the depth m of the modulation. However, m should be as small as possible because the second harmonic distortion ratio ϵ is equal to m. The sound pressure of the secondary wave is also proportional to the square of the initial sound pressure p_0 of the carrier sound and the square of the beam radius a. These values must be as large as possible to obtain adequate sound pressure for practical use.

Since the frequency response of the secondary wave has ω^2 characteristics, an equalizer is required for flat response. Usually, it is quite difficult to produce low-frequency sound because of distortion.

One special feature of this loudspeaker is its very sharp directivity pattern. This loudspeaker can be used as a sound spotlight. Since an acoustic spotlight has never existed in an audible sound region, various uses for this loudspeaker may be anticipated. For example, the sharp directivity would make it possible to speak to one group of people without disturbance to neighboring groups. In a museum or an exhibit, expensive sound barriers between exhibits would be unnecessary.

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Exhibit B

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Parametric Loudspeaker—Characteristics of Acoustic Field and Suitable Modulation of Carrier Ultrasound

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SUMMARY

Due to the inherent nonlinearity of the medium, finite amplitude ultrasound interacts with itself and generates some secondary waves in the sound beam. The parametric loudspeaker, making use of this phenomenon, has a sharp directivity and might be applied to a speech transmission system under the limited environment.

In this paper, a numerical analysis method available for high sound pressure levels, where the nonlinear interactions of ultrasound become greatly active, can be used successfully to theoretically design the parametric loudspeaker. It is reported that the numerical computations agree well with the experiments by a circular aperture projector of radius 21 cm and carrier frequency 27 kHz.

To develop the parametric loudspeaker for practical uses, the problems on harmonic distortions and the physiological effect on human being must be solved. Based on the theoretical prediction, the reasonable solutions for such problems are considered in accordance with appropriate primary wave modulations.

1. Introduction

When two finite amplitude ultrasounds with different frequencies are radiated collinearly, the combination sounds such as the difference frequency wave in addition to the harmonics are generated secondarily in the sound beams due to the nonlinearity of the medium. The secondary waves increase their amplitudes cumulatively as the primary waves propagate, and the amplification process continues until the primary waves decay by the sound absorption and spherical spreading. The generation mechanism is exactly like the end-fire array in antenna theory. Hence, the difference frequency sound has an extremely narrow beam with low frequency. The sidelobes inherent to a directive sound source is suppressed. Such a hypothetical end-fire array in the generation process of the difference frequency is called a parametric acoustic array.

Since Westervelt's concept of the parametric array [1], acoustic characteristics of the array and the application to sound sources have been carried out in underwater acoustics. This is because the conventional transducers can be used directly for radiation of large amplitude primary waves. The parametric array in air has been confirmed experimentally by several authors [2, 3]. Nevertheless, the application researches are not well investigated. The idea of application to an audio loudspeaker is dependent on the development of the ultrasonic device in air [4].

When the parametric array is realized as a loudspeaker (in the following, it is called parametric loudspeaker), the primary frequencies are chosen to be in the ultrasonic frequency range above 20 kHz. At around 40 kHz, ultrasound typically is used. The primary wave which is amplitude-modulated by speech or music signal is demodulated by the nonlinearity of air; that is, the audible sound which corresponds to original signals begins to appear progressively. The parametric array really exists and the beam pattern of the demodulated sound is very narrow.

The present research considers the practical developments of the parametric loudspeaker for a speech transmission system under a limited environment such as a reverberant sound field.

First, we describe the acoustic field

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characteristics of the parametric loudspeaker. Second, we propose an optimum modulation method of primary waves to reduce the ultrasonic radiant power and the harmonic distortions. Third, some experiments using a circular aperture parametric loudspeaker of radius 21 cm and carrier frequency 27 kHz confirms the effectiveness of the present modulation method.

2. Acoustic Characteristics

Since Westervelt's original concept of a parametric array, many researchers have reported the theoretical analyses of the parametric acoustic field. To obtain difference frequency sounds, Muir and Willette have integrated numerically the inhomogeneous wave equation including the beam spreading of primary waves [5]. Moffett and Mellen divided the primary field into plane and spherical wave regions hypothetically [6]. By adding the difference frequency sounds generated in each region, parametric gain is obtained. Although the former is an easily understandable analysis, the integration range has to be determined empirically. In the latter, the difference frequency can be calculated even for the primary waves with large amplitude. However, it describes the nearfield behavior inadequately. On the other hand, Tjotta et al. solved the Khokhlov-Zabolotskaya-Kuznetsov (KZK) equation [7]

$$\frac{\partial}{\partial\tau}\left(\frac{\partial p}{\partial z}-\frac{\beta}{\rho_0 c_0^3}p\frac{\partial p}{\partial\tau}-\frac{b}{2\rho_0 c_0^3}\frac{\partial^2 p}{\partial\tau^2}\right)=\frac{c_0}{2}\nabla_1^2 p \quad (1)$$

by the successive approximation and discussed the nearfield characteristics of the difference frequency in detail [8]. Here, p is the sound pressure, $\tau=t-z/c_0$, $\nabla_1^2=\partial^2/\partial x^2\ \partial^2/\partial y^2$, β, is the nonlinearity parameter, ρ_0 is the medium density, c_0 is the small amplitude sound speed, b is the coefficient related to sound absorption, z is the propagation axis, and x and y are the coordinates orthogonal to the z-axis. Obviously, the linearized and quasi-linearized solutions derived by Tjotta et al. no longer hold when the nonlinear interactions of primary waves are significant.

Since the KZK model equation accurately describes the nonlinear propagation of a directive acoustic beam, we have proposed a new computational method to obtain the difference frequency sound pressure [9]. This method is based on an extension of the scheme of Aanonsen [10], who analyzed numerically the KZK equation for an initially monochromatic wave, to the case of the two-frequency wave excitation. Not only are the primary waves and their harmonics obtained successfully but also all the combination sounds such as the difference frequency.

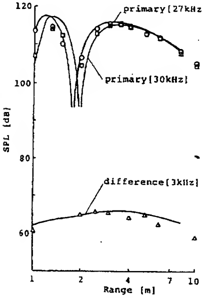

Fig. 1. Axial SPLs of primary and difference frequency waves.

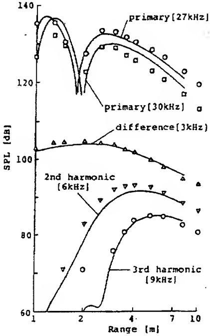

Fig. 2. Axial SPLs of primary, difference frequency and its harmonic waves.

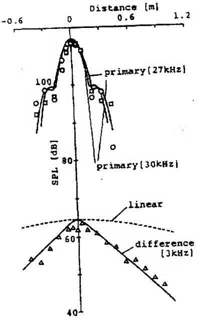

Fig. 3. Beam patterns of primary and difference frequency waves.

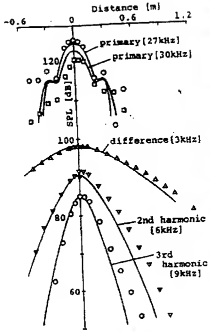

Fig. 4. Beam patterns of primary, difference frequency and its harmonic waves.

Although considerable computation time is required, it is demonstrated that the numerical results agree well with the experiments in air for 5-kHz difference frequency sounds.

In this paper, the same numerical computation is performed to obtain the ~~3-kHz~~ difference frequency. The sound source we used for the experiments consists of 1410 small piezoelectric transducers whose diameter is 1 cm and resonance frequency is 28 kHz. These elements are fixed circularly and regularly on a planar board. The effective radius of the source is 21 cm. Two frequencies for the primary waves are 27 and 30 kHz, respectively. Except that the integration range along the radial direction is 1.33 times greater than the range in the previous paper [9], other numerical parameters are all the same.

Figure 1 shows the axial pressures of the primary and difference frequency waves for rms source pressure level 112 dB. The solid lines denote the numerical computation. Some pressure difference is observed between the computation and experiment for the primary waves near the source and for the difference frequency wave in the far region. However, the overall agreement is quite good. The secondary wave pressure attains the peak around 4 m and then decays gradually with propagation; this curve is a feature of the parametric array [8].

The propagation curves for each spectral component are shown in Fig. 2, thereby increasing the source pressure by 20 dB. Due to the nonlinear attenuation, 27- and 30-kHz waves decay significantly in the region beyond 3 m, and, consequently, the parametric amplification is reduced compared with the data in Fig. 1. The increase of 40 dB near the source is reduced to 33 dB at the range of 8 m. Beyond 3 m, the pressure decays by 6 dB/dd. Therefore, the effective length of the array is restricted by nonlinear attenuation. Further, the difference frequency harmonics at 6 and 9 kHz which neglect amplitude for the low source level are generated abruptly with propagation. Such harmonic generation causes the distortions of the parametric loudspeaker. The experimental pressure difference between two primary waves is larger than the theory. This is probably because the resonant frequency of the transducers tend to decrease with the source pressure.

Figures 3 and 4 show the beam patterns of spectral components at the range of 6 m. For reference, a hypothetical radiation pattern of 3-kHz sound that would be generated linearly by the same radius source (piston

source) is plotted with a dotted line. The directivity of the parametric array is clearly sharper than that of the piston source; at 1 m from the acoustic axis the pressure level decreases by more than 20 dB for the array, while the reduction is only 4.3 dB for the piston source. Although the primary ultrasounds decay greatly at a distance 20 cm off the axis, the difference frequency sound decreases by only several decibels.

As seen in Fig. 4, the beam pattern becomes slightly broad as the source level increases. Nevertheless, the parametric array provides a more narrow directivity.

3. Modulation Systems

Consider the primary wave modulation method when the parametric array is used as an audio loudspeaker. It is assumed that the primary wave is highly collimated and the period of the primary is much shorter than that of the difference frequency for theoretical convenience. These assumptions lead to the following equation that satisfies the secondary wave pressure P_D :

$$\nabla_\perp^2 P_D - \frac{2}{c_0}\frac{\partial^2 P_D}{\partial_\perp \partial_z} = -\frac{\beta}{\rho_0 c_0^4}\frac{\partial^2}{\partial\tau^2}\langle p_\omega^2\rangle \qquad (2)$$

from Eq. (1) [11]. Here, p_ω is the primary pressure and $\langle\ \rangle$ implies the averaging over a time longer than the period of the primary. The absorption of the secondary wave is neglected. When the initial waveform is

$$p_\omega(t\,;\,z=0) = p_0 f(t)\sin\omega t \qquad (3)$$

Merklinger gives the axial pressure as follows [12]:

$$p_\omega = \frac{p_0 f(\tau)\sin\omega\tau}{[1+\{\beta\omega p_0 f(\tau)(1-e^{-2\alpha z})/4\alpha\rho_0 c_0^3\}^2]^{1/2}} \qquad (4)$$

where p_0 is the source pressure amplitude, and $f(\tau)$, ω and α are the envelope, angular frequency and absorption coefficient of the carrier wave, respectively. Substituting Eq. (4) into Eq. (2),

$$\nabla_\perp^2 p - \frac{2}{c_0}\frac{\partial^2 p}{\partial\tau\partial z} = -\rho_0\frac{\partial q}{\partial\tau}$$
$$q = \frac{\beta p_0^2}{2\rho_0^2 c_0^4}\frac{\partial}{\partial\tau}\left[\frac{f^2(\tau)}{1+\{\beta\omega p_0 f(\tau)(1-e^{-2\alpha z})/4\alpha\rho_0 c_0^3\}^2}\right] \qquad (5)$$

Note that the notation of P_D is replaced with p. The solution of Eq. (5) on the axis far away from the loudspeaker is given by

$$p(\tau;z) = \frac{a^2 p_0}{4\omega c_0 z}\frac{\partial^2}{\partial\tau^2}\left[f(\tau)\tan^{-1}\left\{\frac{\beta\omega p_0 f(\tau)}{4\alpha\rho_0 c_0^3}\right\}\right] \qquad (6)$$

Merklinger has already derived the same equation as Eq. (6) [12]. In Eq. (6), if $p_0 \ll 4\alpha\rho_0 c_0^3/\beta\omega$

$$p(\tau,z) \propto \frac{\partial^2}{\partial\tau^2} f^2(\tau) \qquad (7)$$

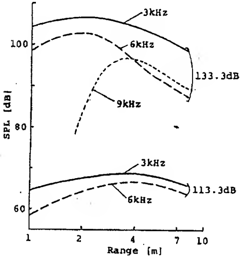

Fig. 5. Computed axial SPLs of difference frequency and its harmonic components for 100 percent AM excitation.

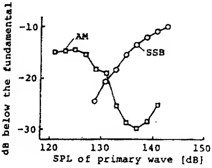

Fig. 6. Relative level of difference frequency second harmonic component as a function of primary source level.

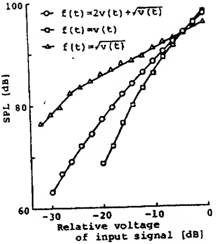

Fig. 7. SPLs of the difference frequency as a function of input signal $s(t)$.

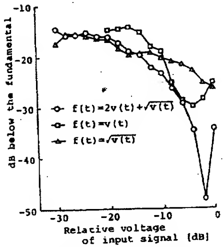

Fig. 8. Relative level of difference frequency second harmonic component as a function of input signal $s(t)$.

On the other hand, if $p_0 \gg 4a\rho_0 c_0^3/\beta\omega$, then

$$p(r, z) \propto \frac{\partial^2}{\partial \tau^2}|f(\tau)| \tag{8}$$

It is clear that the difference frequency sound is proportional to the square of the carrier when the primary pressure is low, and to the envelope itself when the pressure is increased. Hence, to obtain the difference frequency sound with harmonic distortion as little as possible, SSB (single sideband modulation) is advantageous for the small amplitude primary wave, and AM is a profitable modulation for the large amplitude.

Figure 5 shows the difference frequency sound computed with the condition of 100 percent AM primary wave, where the carrier frequency is 27 kHz and the signal frequency is 3 kHz. The source pressure p_0 is set so that the acoustic radiation power is identical to the value in Figs. 1 and 2. The pressure level differences of the difference frequency sound and its second harmonic are almost the same at a near distance for both the source pressures of 113.3 and 133.3 dB. The difference becomes larger for 133.3 dB in the range beyond 1.5 m. Therefore, when the primary pressure level is high, the harmonic distortions of the difference frequency are reduced by means of AM excitation.

To confirm the forementioned theoretical prediction, the level difference was measured for AM and SSB excitations and is shown in Fig. 6. The sound source is the parametric loudspeaker described in section 2. The carrier frequency and signal frequency are 27.4 and 3 kHz, respectively, and the modulation index is 100 percent (the electric signals of 27.4 and 24.4 kHz are the same levels in SSB). The receiving point is at the range of 6 m. It is found that the source pressure reverses the trend of the level differences at around 130 dB.

Instead of the conventional AM method, we have proposed an envelope modulation which controls the radiant power of ultrasound [13], i.e., the carrier is modulated by the signal $f(t)$ which is sum of the signal $s(t)$ and its envelope $e(t)$:

$$f(t)=\{e(t)+s(t)\}^{1/2} \tag{9}$$

By means of this modulation, the average electric power was decreased by suppression of primary waves in the absence of modulation signals. Consequently, the unnecessary exposure of ultrasound to physiological sense is definitely reduced. For instance, in a news program, the power was reduced to 36 percent compared with the AM. However, the envelope modulation has two drawbacks: one of them is the generation of harmonic distortion; and the other is the degradation of linearity of the difference frequency sound. To remove these drawbacks, a new type of modulation will be introduced. Depending on source levels, the modulation signal is changed gradually from Eq. (9) to

$$f(t)=e(t)+s(t) \tag{10}$$

Hence, let

$$v(t)=e(t)+s(t) \tag{11}$$

The modulation is realized by the following equation:

$$f(t)=x\cdot v(t)+v(t)^{1/2} \tag{12}$$

where x is an empirically determined constant. The input-output relation of the difference frequency sound is given in Fig. 7. Under the present experiments the linearity is the best when $x = 2$, thus only the data for this case are presented. In comparison with other modulations by Eq. (9) and (10), the present method has the superior linearity, and much less harmonic distortion as shown in Fig. 8.

4. Conclusions

The theory and experiment are done for the acoustic field characteristics of a parametric loudspeaker. The numerical calculation method by Aanonsen is extended to the case where a two-frequency wave is radiated initially. It has been shown clearly that the measured data of not only the difference frequency but also the primary and combination sounds are in fairly good agreement with the theoretical prediction.

As a driving method for the parametric loudspeaker, a new type of envelope modulation is proposed in which the electric power consumption and the amount of ultrasonic exposure as well as reduced compared with the conventional AM excitation. The theoretical description of the modulation is made and its effectiveness is confirmed by the experiments. The exposure problem is an important topic and has to be considered in the use of a loudspeaker. This problem will be discussed later with application examples of the parametric loudspeaker.

Acknowledgment. The authors are indebted to a number of individuals in the course of this work, especially the members of the Nonlinear Acoustic Society of Japan. Mr. N. Hamada, graduate student, assisted in experiments and Mr. T. Tanaka of Matsushita Electric has been instrumental in the work on the ultrasonic transducer. This work was supported in part by a Grant-in-Aid for Scientific Research, Ministry of Education (No. 61850058).

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Exhibit C

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Research Notes

Suitable Modulation of the Carrier Ultrasound for a Parametric Loudspeaker

Zweckmäßige Modulation des Ultraschallträgers für einen parametrischen Lautsprecher

Modulation de la porteuse ultrasonore en vue de réaliser un haut-parleur paramétrique

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Introduction

A parametric loudspeaker, the directivity of which is very sharp, uses the self-demodulation effect of finite amplitude ultrasound being amplitude-modulated by audio signals [1]. The constant amplitude carrier wave is caused to interact non-linearly with the side-band waves to produce audible sounds in air. In order to develop the parametric loudspeaker for practical purposes such as speech transmission to a limited area in a reverberent hall, some important problems have to be settled [2].

In the present short article, the emphasis is on the practical modulation of the carrier ultrasound, especially regard to the effective reduction of the radiant power when working the loudspeaker.

Modulation methods

Let an amplitude-modulated ultrasonic wave:

$$p = p_0 \, f(t) \sin \omega_0 t \tag{1}$$

be radiated from a parametric loudspeaker in free space. Here p is the primary sound pressure with initial value p_0, and ω_0 is the angular frequency of the carrier wave. The function $f(t)$ varies much more slowly than the carrier $\sin \omega_0 t$ does. If the ultrasound beam is fully collimated, the axial sound pressure of the parametric signal in the farfield region is proportional to the second derivative of $f(t)^2$ with respect to time, provided non-linear absorption is not significant [3]:

$$p_s = \text{const} \times \frac{\partial^2}{\partial t^2} f^2(\tau), \tag{2}$$

where $\tau = t - z/c_0$, c_0 is the speed of small amplitude sound, and z is the distance from the loudspeaker. When $f(t)$ takes the form linearly related to the audio signal $s(t)$ as

$$f(t) = 1 + m\, s(t), \qquad \text{DSB modulation} \tag{3}$$

p_s becomes

$$p_s = \text{const} \times \frac{\partial^2}{\partial t^2} \{2m\, s(\tau) + m^2 s^2(\tau)\}, \tag{4}$$

where m is a modulation factor which satisfies the condition $|m\, s(t)| \leq 1$ [1]. In eq. (4) the first term of the right-hand side is the demodulated sound which comes from the interaction between the carrier and side-band waves. On the other hand the second term, whose magnitude increases with m squared, is an undesirable sound containing distortion due to the interaction between the upper and lower side-band waves. The higher the pressure level of the audio sound increases, the more the sound waves distort. To reduce the distortion, one of the authors and his coworkers have proposed a modified amplitude modulation [2]:

$$f(t) = \{1 + m\, s(t)\}^{1/2}. \tag{5}$$

Using the square-rooted envelope in this way, sound without distortion can theoretically be obtained. In general the bandwidth of the primary signal widens because of the non-linear transform of the envelope. Hence every transducer which is an element of the loudspeaker requires a wide-band frequency response characteristic. It has been verified that the harmonic distortions are less than those that the double-side band (DSB) and single-side band (SSB) excitations generate [4].

Although any of the modulations described above are direct and simple, carrier ultrasound of finite amplitude is always radiated, even if the audio signal is very small or almost zero. Consequently, attention to the problems not only of electric power loss, but also of the physiological effects on human beings has to be paid. A new type of modulation is presented here in order to reduce the average radiant power of ultrasound as effectively as possible.

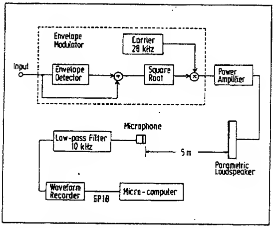

Fig. 1. Block diagram of experimental arrangement.

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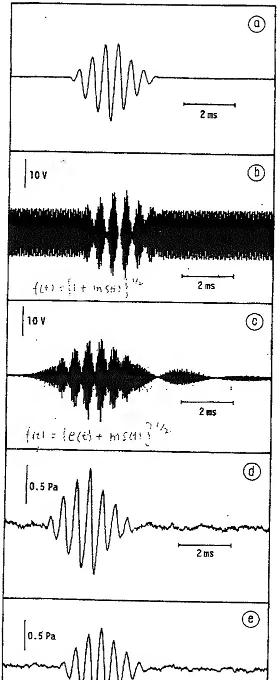

Fig. 2. Waveforms obtained in the process of the experiment. a) Original tone-burst signal of 2 kHz, b) waveform of driving voltage (DSB modulation), c) waveform of driving voltage (envelope modulation), d) parametric sound by DSB modulation, e) parametric sound by envelope modulation. The parametric sounds are received at a range of 5 m.

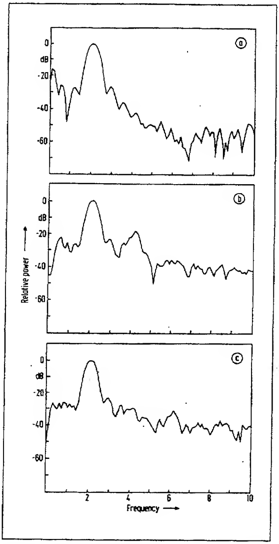

Fig. 3. Power spectra of the original and parametric signals. a) Original signal, b) parametric signal (DSB modulation), c) parametric signal (envelope modulation).

Audio signals such as speech vary dynamically in time. Now let the envelope $e(t)$ be exactly extracted from the signal. When the function $f(t)$ has the following form

$$f(t) = \{e(t) + m\, s(t)\}^{1/2}, \tag{6}$$

the demodulated sound consists of two signals; one of them is the original signal $s(t)$, and the other is the envelope signal $e(t)$. No sooner is $s(t)$ present than the carrier wave is radiated; the audio signal controls directly the ultrasonic radiant power. As the envelope contains only low frequency components, and the magnitude of demodulated sound is related to the square of the frequency, the envelope sound is too feeble to be heard. Hereafter this will be referred to as envelope modulation for brevity.

Experiments and discussion

To demonstrate the validity of the envelope modulation, the experiments were performed with a parametric loudspeaker of rectangular aperture (44 × 50) cm² in size, consisting of about 2000 small PZT bimorph transducers of resonance frequency 28 kHz. A block diagram of the experimental arrangement is given in Fig. 1. The modulator is compactly constructed in accordance with eq. (6), and includes a low-pass filter which removes the high frequency components of the envelope above 200 Hz.

Fig. 2 illustrates the waveforms obtained in the process of the experimental procedure. Fig. 2a is an original tone-burst signal of 2 kHz. The factor m is kept at unity. Waveforms of the voltage driving the loudspeaker for the excitations of DBS modulation and envelope modulation are given in Figs. 2b and 2c, respectively. The corresponding waveforms of parametric sounds received by a 1/4-inch condenser microphone at a range of 5 m are shown in Figs. 2d and 2e. As the carrier ultrasound pressure is higher than the parametric pressure, a precise low-pass filter with cut-off frequency 10 kHz attenuates greatly the carrier and its harmonic components. In order to make a clear comparison between the original and parametric signals, the power spectra are shown in Fig. 3. It seems to be apparent that the spectrum from the envelope modulation method closely resembles the original spectrum. The electric average power consumed by working the parametric loudspeaker was measured under the condition of the same operations for both excitations. A news programme spoken by a male announcer was used for the speech transmission test. Table I summarises the results of the average power. Without degradation of tone quality the power used by the envelope modulation method has been reduced to about one-third of that used by the DSB modulation.

Table 1. Average powers consumed in working the parametric loudspeaker.

News program (84 s)	Average Power (DSB) W	(envelope) W	Power Reduction %
	390	140	36

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A Statistical Method of Evaluating the Sound Insulation Effect of a Single Wall

Eine statistische Methode zur Auswertung des Schallisolationseffekts einer Einfachwand

Une méthode statistique pour évaluer l'isolation phonique procurée par un mur simple

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1. Introduction

As is well known, statistics such as the L_x noise level (e.g. L_5, L_{10} and L_{50}), defined as the $(100-\alpha)$ percentage point of the noise level distribution, and also the lower order energy statistics such as L_{eq}, are very important in actual noise evaluation and control problems. Accordingly, it is necessary to establish a systematic method for evaluating the effect of sound insulation systems on the widely-used noise evaluation index L_α. A statistical method of evaluating sound insulation systems with random incident waves has been investigated previously [1–3]. In these papers, however, the effect of the sound insulation system on the decrease of L_x was not clearly shown in the functional form of the system parameters (such as the surface density of the wall, the angle of incidence etc.).

In this paper, the above evaluation problem is considered from a different viewpoint. By using the idea of a shaping filter [4], a statistical treatment of the probability density function (pdf) is first proposed for the transmitted sound pressure in the case when a Gaussian type of random sound pressure wave with an arbitrary power spectral density is passed through an arbitrary linear sound insulation system. Next, by using the above general theory, the sound insulation effect of a single wall is considered. As a result, an explicit expression for the sound insulation effect, defined as the difference between the L_x of the incident sound intensity and that of the transmitted sound intensity, is derived as a function of the system parameters. Finally, the validity of the theoretical evaluation method is confirmed by a simulation experiment.

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2. The probability density function for a transmitted sound pressure wave

When an incident sound pressure wave, $x(t)$, is passed through a sound insulation system, the transmitted sound